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# 1/77

## Request for grant of a patent

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17 JAN 2003

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1. Your reference

P163-GB

2. Patent application number
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0301093.1

3. Full name, address and postcode of the or of each applicant *(underline all surnames)*

1... Limited
St John's Innovation Centre
Cowley Road
Cambridge CB4 0WS

Patents ADP number *(if you know it)*

If the applicant is a corporate body, give the country/state of its incorporation

England 8113870001

4. Title of the invention

SET-UP METHOD FOR ARRAY-TYPE SOUND SYSTEMS

5. Name of your agent *(if you have one)*

Akram K. Mirza

"Address for service" in the United Kingdom to which all correspondence should be sent *(including the postcode)*

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Cambridge CB4 0WS 8113870001

Patents ADP number *(if you know it)*

6. If you are declaring priority from one or more earlier patent applications, give the country and the date of filing of the or of each of these earlier applications and *(if you know it)* the or each application number

| Country | Priority application number *(if you know it)* | Date of filing *(day / month / year)* |
| --- | --- | --- |

7. If this application is divided or otherwise derived from an earlier UK application, give the number and the filing date of the earlier application

| Number of earlier application | Date of filing *(day / month / year)* |
| --- | --- |

8. Is a statement of inventorship and of right to grant of a patent required in support of this request? *(Answer 'Yes' if:*
*a) any applicant named in part 3 is not an inventor, or*
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*See note (d))*

YES

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| Claim(s) | 3 |
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| | |
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| Priority documents | |
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| Statement of inventorship and right to grant of a patent *(Patents Form 7/77)* | |
| Request for preliminary examination and search *(Patents Form 9/77)* | |
| Request for substantive examination *(Patents Form 10/77)* | |
| Any other documents *(please specify)* | |

11. I/We request the grant of a patent on the basis of this application.

Signature Date 17-JAN-2003

12. Name and daytime telephone number of person to contact in the United Kingdom Akram K. Mirza 01223-422290

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P163-GB

# SET-UP METHOD FOR ARRAY-TYPE SOUND SYSTEM

## FIELD OF THE INVENTION

This invention relates to a device including an array of electro-acoustic transducers capable of receiving a multi-channel audio input signal and to produce independently steerable and focusable beams of audible sound, at a level suitable for home entertainment or professional sound reproduction applications. More specifically, the invention relates to the methods and systems for configuring such a device.

## BACKGROUND OF THE INVENTION

The commonly-owned International Patent applications no. WO-0123104 and WO-02078388 describe an array of transducers and their use to achieve a variety of effects. They describe methods and apparatus for taking an input signal, replicating it a number of times and modifying each of the replicas before routing them to respective output transducers such that a desired sound field is created. This sound field may comprise a directed, steerable beam, focussed beam or a simulated origin. The methods and apparatus of the above and other related applications is referred to in the following as "sound projector" technology.

Conventional surround-sound is generated by placing loudspeaker at the appropriate position surrounding the listener's position or "sweet-spot". Typically, a surround-sound system employs a left, centre and right speaker located in the front halfspace and two rear speakers in the rear halfspace. The terms "front", "left", "centre", "right" and "rear" are used relative to the listener's position and orientation.

A surround-sound system decodes the input audio information and uses the decoded information to distribute the signal among different channels with each channel usually being emitted through one or loudspeaker or combination of two speakers.

In the commonly-owned published international patent applications no. WO-0123104 and WO-02078388 the sound projector generates the surround-sound environment by emitting beams of sound each representing one of the above channels and reflecting such beams from surfaces such as ceiling and walls back to the listener. The listener perceives the sound beam as if emitted from an acoustic mirror image of a source located at or behind the spot where the last refection took place.

Whereas Sound Projector systems that use the reflections of acoustic beams can be installed by trained installers and closely guided users, there remains a desire to facilitate the set-up procedure for less-trained personnel or the average end user.

The problems associated with the setting up of a sound projector are not related to certain known methods aiming at partial or total wavefield reconstruction. In the latter methods, it is attempted to record a full wavefield at the listener's position. For reproduction a number of loudspeakers are controlled in a manner that closest approximates the desired wavefield at the desired position. Even though these methods are inherently recording reflections from the various reflectors in a room or concert hall, no attempt is made to infer from these recordings control parameters for a sound projector. In essence, the wavefield reconstruction methods are "ignorant" as to the actual room geometry and therefore

not applicable to the control problem underlying the present invention.

## SUMMARY OF THE INVENTION

The present invention proposes the use of one or a combination of two or more of the following methods to facilitate the installation of a sound projector:

A first approach is use a set-up guide in form of an electronic medium such as CDROM or DVD, or a printed manual manual, preferably supported by a video display. The user is asked a series of questions, including details of:
- The mounting position of the sound projector;
- The shape and dimensions of the room; or
- The distance to the listening position from the Sound Projector.

This can either be done through a series of open questions, as in an expert system, or by offering a limited choice of likely answer combinations, together with illustrations to aid clarity.

From this information, a few beam directions for each channel can be pre-selected and stored, for example in form of a list. The sound projector system can then produce short bursts of band-limited noise, cycling repeatedly through each of these directions. For each direction the user is then asked to select a (subjective) best beam direction, for example by activating a button. This step can be repeated iteratively to refine the choice.

Without making use of a microphone, the user might then be asked to select from a menu the type of surface on each wall and on the ceiling. This selection, together with the steering angles as established in the previous step, could be used to

derive an approximate equalisation curve. Delay and level matching between channels could be performed using a similar iterative method.

In variants of the above method where a microphone is used attached to an input socket of the sound projector, a more automated approach can be taken. With an omni-directional microphone positioned at the main listening position, the impulse response can be measured automatically for a large number of beam angles, and a set of local optima, at which there are clear, loud reflections, can be found. This list can be refined by making further automated measurements with the microphone positioned in other parts of the listening area. Thereafter the best beam angles may be assigned to each channel either by asking the user to specify the direction from which each beam appears to come, or by asking questions about the geometry and deducing the beam paths. Asking the user some preliminary questions before taking measurements will allow the search area, and hence time, to be reduced.

A faster, automated approach includes the step of measuring the impulse responses between a number of single transducers on the panel and a microphone at the listening position. By decomposing the measured impulse responses into individual reflections and using a fuzzy clustering or other suitable algorithm, it is possible to deduce the position and orientation of the key reflective surfaces in the room, including the ceiling and side walls. The position of the microphone (and hence the listening position) relative to the Sound Projector can also be found accurately and automatically.

These and other aspects of inventions will be apparent from the following detailed description of non-limitative examples and drawings.

## BRIEF DESCRIPTION OF THE DRAWINGS

FIG. 1 is a schematic drawing of a typical set-up of a sound projector system in accordance with the present invention;

FIG. 2 is a graph showing registered responses to a signal emitted by a transducer of the sound projector system;

FIG. 3 is a modeled impulse response for a idealized room;

FIG. 4 shows results of cluster analysis performed on registered responses to signals emitted from different transducers of the sound projector system;

FIG. 5 summarizes the general steps of a method in accordance with the invention.

## DETAILED DESCRIPTION

The present invention is best illustrated in connection with a digital sound projector as described in the co-owned applications no. WO-0123104 and WO-02078388.

Referring to FIG.1, a digital loudspeaker system or sound projector **10** includes an array of transducers or loudspeakers **11** that is controlled such that audio input signals are emitted as a beam or beams of sound **12-1, 12-2**. The beams of sound **12-1, 12-2** can be directed into - within limits - arbitrary directions within the half-space in front of the array. By making use of carefully chosen reflection paths, a listener **13** will perceive a sound beam emitted by the array as if originating from the location of its last reflection or - more precisely- from an image of the array as reflected by the wall, not unlike a mirror image.

In FIG. 1, two sound beams **12-1** and **12-2** are shown. The first beam **12-1** is directed onto a sidewall **161**, which may be part of a room, and reflected in the direction of the listener **13**. The listener perceives this beam as originating from an image of the array located at, behind or in front of the reflection spot **17**, thus from the right. The second beam **12-2**, indicated by dashed lines, undergoes two reflections before reaching the listener **13**. However, as the last reflection happens in a rear corner, the listener will perceive the sound as if emitted from a source behind him or her.

Whilst there are many uses to which a sound projector could be put, it is particularly advantageous in replacing conventional surround-sound systems employing several separate loudspeakers placed at different locations around a listener's position. The digital sound projector, by generating beams for each channel of the surround-sound audio signal and steering those beams into the appropriate directions, creates a true surround-sound at the listener position without further loudspeakers or additional wiring.

The components of a digital sound projector system are described in the above referenced published International patent applications no. WO-0123104 and WO-02078388 and, hence, reference is made to those applications.

In the following the steps leading to the automated identification of reflecting surfaces, such as side-wall **161** in FIG.1, in a room with a sound projector.

For the subsequent method it is assumed that the sound projector is centred on the origin and lies in the yz plane where the positive y axis points to the listeners' right and the positive z axis points upwards; the positive x axis points towards the listeners.

The microphone is modeled by a point in space and is assumed to be omnidirectional. Under the assumption that the reflective surfaces are planar, the system can be thought of as an array of microphone "images" in space, with each image representing a different sound path from the transducer array to the microphone. The speed of sound c is assumed to be known, i.e. constant, throughout, so distances and travel-times are interchangeable.

Given a microphone located at (xmic; ymic; zmic) and a transducer located at (0; yi; zi), the path distance to the microphone is

[1] di = (xmic^2 + (ymic-yi)^2 + (zmic-zi)^2)^(1/2),

which can be rewritten as the equation of a two-sheeted hyperboloid in (di; yi; zi) space as follows:

[2] di^2 - (ymic-yi)^2 - (zmic-zi)^2 = xmic^2

The "^" notation indicates an exponent.

At a listening distance of 3m, the effects of curvature of the hyperboloid are negligible over the available range of yi and zi and it can be considered either planar or hyperboloidal

To measure a impulse response, a single transducer is driven with a known signal, for example five repeats of a maximum length sequence of 2^18- 1 bits. At a sampling rate of 48kHz this sequence lasts 5.46 seconds.

A recording is taken using the omnidirectional microphone within the listening area. The recording is then filtered by convolving it with the time-reversed original sequence and the

correlation is calculated by adding the absolute values of the convolved signal at each repeat of the sequence, to improve the signal-to-noise ratio.

The above impulse measurement is performed for several different transducers in the array of the sound projector. Using multiple sufficiently uncorrelated sequences can shorten the time for these measurements. With such sequences it is possible to measure the impulse response from more than one transducer simultaneously.

In order to test the following algorithms, a listening room was set up with a Mk 5a DSP and an omnidirectional microphone at roughly (4:0; 0:0; 0:6), and six repeats of a maximum length sequences (MLS) of 2^18-1 bits was sent at 48kHz to individual transducers by selecting them from the on-screen display. Transducers 0, 9, 33, 57, 66, 90, 114, 123, 133, 159, 193, 219, and 252 were used (13 in all) of the 256 transducer array of the sound projector, forming a roughly evenly spaced grid across the surface of the DSP including transducers at "extreme" position, such as the centre or the edges. The microphone response was recorded to 48kHz WAV-format files for analysis.

The time-reversed original MLS was convolved with the response from each transducer in turn and the resulting impulse response normalized by finding the first major peak (corresponding to the direct path) and shifting the time origin so this peak was at t = 0, then scaling the data so that the maximum impulse had height 1. The time shift alleviates the need to accurately synchronize the signals.

A segment of the impulse response of transducer 0 (in the top-left corner of the array) is shown in FIG. 2. The graph shows the relative strength of the reflected signal relative to the

travel path length as calculated from the arrival time. Several peaks (above -20 dB) are identifiable in the graph, for example the peaks at 0.4m, 1.2m, 3.0m, 3.7m and 4.4m.

Before attempting to associate these peaks with reflectors in a room, a model of a perfectly reflecting room is illustrated in FIG. 3.

FIG. 3 is a graph of the `perfect' impulse response of a room with walls 2.5m either side of the sound projector, a rear wall 8m in front of it and a ceiling 1.5m above it, as heard from a point at (4; 0; 0). As the signal is reflected from reflecting surfaces the microphone measures an refection image of that surface in accordance with the path or delay values from equations [1] or [2]. The direct path and reflections from the ceiling correspond to the first two surface images **311**, **312**, and the next four intermingled arrivals **313** correspond to the reflections from the sidewalls with and without the ceiling, respectively. Other later arrivals **314**, **315** represent reflections from the rear wall or multiple reflections.

Using the model of FIG. 3, a plausible interpretation of some of the major peaks of FIG. 2 can be given. Table 1 below lists these interpretations.

TABLE 1.

| Distance (m) | Likely source |
|---|---|
| 0 | Direct path from transducer to microphone |
| 0.4 | Reaction from coffee table |
| 1.2 | Reaction from ceiling |
| 3.0, 3.7, 4.4 | Reaction from side walls with/without ceiling. |

The algorithms detailed below are concerned with performing this analysis automatically without prior knowledge of the shape of the room or its contents and thus identifying suitable reflecting surfaces and the orientation with respect to the sound projector.

After or while measuring the impulse response from several transducers located at different positions spread across the array the data is searched for arrivals that indicate the presence of reflecting surfaces in the listening room.

In the present example the search method is making use of an algorithm that identifies clusters in the data.

In order to improve the performance of the clustering algorithm, it is useful to perform a preclustering step to remove a large quantity of noise from the data and to remove large spaces devoid of clusters. In the case of FIG. 2, preclusters were selected within the following ranges of minimum level in dB and minimum and maximum distance in meters: precluster 1 (-15, 0, 2); precluster 2 (-18, 2.8, 4.5), and precluster 3 (-23, 9, 11).

Once the data has been separated roughly into a noise cluster and a number of clusters which potentially contain impulses from reflections, a modified version of the fuzzy c-varieties (FCV) algorithm described for example in James C. Bezdek, "Pattern Recognition with Fuzzy Objective Function Algorithms", Plenum Press, New York 1981, is applied to the data to seek out planes of strong correlation. The 'fuzziness' of the FCV algorithm comes from a notion of fuzzy sets: the ith data point is a member of the kth fuzzy cluster to some degree, called the degree of membership and denoted U(ik). The matrix U is known as the membership matrix.

The FCV algorithm relies on the notion of a cluster "prototype", a description of the position and shape of each cluster. It proceeds by iteratively designing prototypes for the clusters using the membership matrix as a measure of the importance of each point in the cluster, then by reassigning membership values based on some measure of the distance of each point from the cluster prototype.

The algorithm is modified to be robust against noise by including a "noise" cluster which is a constant distance from each point. Points which are not otherwise assigned to "true" clusters are classified as noise and do not affect the final clusters. This modified algorithm is referred to as "robust FCV" or RFCV.

It is common when running the algorithm that it will converge to a local optimum which is not optimal enough, in the sense that it does not correspond to a cluster representing a reflection. This issue is corrected by waiting for the rate of convergence to drop low enough that further large changes become unlikely (typically a change-per-iteration of 10^-3) and to check the validity of the cluster. If it is deemed to be invalid then the next step involves a jump to a randomly chosen point elsewhere in the search space.

The original FCV algorithm relies on fixing the number of clusters before running the algorithm. A fortunate side-effect of the robustness of the modified algorithm is that if too few clusters are selected it will normally be successful in finding as many clusters as were requested. Thus a good method for using this algorithm is to search for a single cluster, then a second cluster, and continue increasing the number of clusters, preserving the membership matrix at each step, until no more clusters can be found.

Another parameter to be chosen in the algorithm is the fuzziness degree, m, which is a number in the range between 1 and infinity. The value m = 2 is commonly used as a balance between hard clustering (m ->1) and overfuzziness (m -> infinity) and has been successfully used in this example.

The number of clusters c is initially unknown, but it must be specified when running the RFCV algorithm. One way of how to discover the correct value of c is to try the algorithm for each c up to a reasonable cmax. In its non-robust form and with noise-free data the algorithm will successfully pick out c clusters when c clusters are present. If there are more or fewer than c clusters present, at least one of the clusters that the algorithm finds will fail to pass tests of validity which gives a clear indication as to which value of c is correct.

The robust version performs better when there are more than c clusters present: it finds c clusters and classifies any others as noise. This improvement in performance comes at the expense of having less indication which value of c is truly correct. This problem can be resolved by using an incremental approach, such as follows:

1. Run the algorithm with c = 1 and without specifying the initial membership matrix U0 of the algorithm so that the initial prototype is randomly generated.

2. Repeat the following steps until the algorithm returns fewer than c prototypes:

2.1 Increment c and set U0 to be the final membership matrix of the preceding step, including the membership values into the "noise" cluster.

2.2 Rerun the algorithm.

This method has a number of advantages. Firstly, the algorithm never runs with fewer than c - 1 clusters, so the wait for extraneous prototypes to be deleted is minimized. Secondly, the starting point of each run is better than a randomly chosen one, since c - 1 of the clusters have been found and the remaining data belongs to the remaining prototype(s).

FIG. 4 shows the results of applying the incremental RFCV algorithm on the second precluster of FIG. 2 using c = 1 (FIG. 4A) and c= 2, … 5 (FIGs. 4B, ….4E, respectively.). In the case of c=3 (FIG. 4C) the method converges onto an artifact. As the number of clusters is further increased to c=4 and c=5 (FIGs. 4D, E) this cluster disappears and the four correctly recognized reflectors are recognized in the data. No further cluster is identified. The clusters are indicated by planes **413** drawn into the data space, which in turn is indicated by black dots **400** representing the impulse response of the microphone to the emitted sequences.

As in an automated set-up procedure the microphone position may be an unknown, any cluster identified according to the steps above, can be used to solve with standard algebraic methods equation [2] for the microphone position xmic, ymic and zmic.

With the microphone position and the distance and orientation of images of the transducer array known enough information is known about the room configuration to direct beams at the listeners from a variety of angles. This is done be reversing the path of the acoustic signal and directing a sound beam at each microphone image.

However, it is necessary to deduce the direction from which the beam appears to arrive at the listener.

One way of making this deduction is to decide which walls the beam is being reflected in order to arrive at the microphone. If this decision is to be made automatically then it can be for most cases assumed that the walls are all flat and reflective over their whole surfaces. This implicitly means that the secondary reflection of surfaces A and B arrives at the microphone later than the primary reflected signals from surface A and from surface B, which permits the following algorithm:

1. Start by initializing an empty list of walls.

2. Take each microphone image in order of their distances from the DSP and search through all combinations of walls in the list to see if any composition of reflections in those walls could result in the microphone image being in the right place.

3. If such a combination does not exist then this microphone image is formed by a primary reflection in an as-yet-undiscovered wall. This wall is the perpendicular bisector of the line segment from the microphone image to the real microphone. Add the new wall to the list.

A more robust way of comprises the use of multiple microphones or one microphone positioned at two or more different locations during the measurement and determine the perceived beam direction directly.

Using an arrangement with 4 microphones in a tetrahedral arrangement and after having determined the positions of images of each of the microphones individually they can be grouped into images of the original tetrahedron which will

fully specify the perceived beam direction. If the walls are planar then the transformation mapping the real tetrahedron to its image will be an isometry and its inverse equivalently maps the sound projector to its perceived position from the listener's point of view.

Using less than four microphones results in an increase of uncertainty in the direction of the arrival. However in some case it is possible to use reasonable constraints, for example, such as that wall are vertical etc, to reduce this uncertainty.

The problem of scanning for a microphone image is a 2-dimensional search problem. It can be reduced to two consecutive 1-dimensional search problems using the beam projectors ability to generate various beam patterns. For example it is feasible to vary the beam shape to a tall, narrow shape and scanning horizontally, and then use a standard point-focused beam to scan vertically.

With a normal point-focused beam the wavefront of the impulse is designed to be spherical, centered on the focal point. If the sphere were replaced with an ellipsoid, stretched in the vertical direction, then the beam will become defocused in the vertical direction and form a tall narrow shape.

Alternatively, it is possible to form a tall narrow beam by using two beams focused at two points in space above one another and the same distance away from the sound projector. This is due to the abrupt change of phase between sidelobes and the large size of the main beam in comparison with the sidelobes.

The general steps of the above-described method are summarized in FIG. 5.

## CLAIMS

1. A set-up method for a loudspeaker system capable of generating at least one directed beam of audio sound, comprising the steps of
   - emitting a series of signals from the loudspeaker system into a room including a listener's position; ;
   - registering said series and its reflections at one or more locations within said room;
   - evaluating said registered series to determine a first set of values for directing parameters for said beam; and
   - using said directing parameters to direct said beam into the desired direction.

2. The method of claim 1 wherein the series of signals are emitted from less than the full number of electro-acoustic transducers in the loudspeaker system.

3. The method of claim 1 wherein the series of signals are emitted from single electro-acoustic transducers in the loudspeaker system.

4. The method of claim 3 wherein different series of signals are simultaneously emitted from different electro-acoustic transducers.

5. The method of claim 4 wherein the different electro-acoustic transducers are located at an edge position and/or the centre of the transducer array.

6. The method of claim 1 wherein the series of signals are emitted as spatially-constrained beams of sound to a range of directions.

7. The method of claim 6 wherein the spatially-constrained beams of sound are laterally constrained to form narrow vertical beams.

8. The method of claim 6 wherein the spatially-constrained beams of sound are laterally and vertically constrained to form narrow point or ellipsoidal beams.

9 The method of claim 1 wherein the registering step includes the step of positioning one or more microphones and recording the series of signals using said microphones.

10 The method of claim 9 wherein the one and more microphones are arranged in a tetrahedral configuration.

11. The method of claim 1 wherein the evaluating step includes the step of determining the listener's position relative to the location of the loudspeaker system.

12. The method of claim 1 wherein the evaluating step includes the step of identifying multiple paths to the listener's position.

13. The method of claim 1 wherein the evaluating step includes the step of identifying multiple acoustic paths to the listener's position and assigning different audio channels to different paths.

14. The method of claim 1 wherein the evaluating step includes the step of identifying clusters of reflections in the registered series.

15. The method of claim 1 using pre-known data relating to the geometry of the room to exclude beam directions.

16. The method of claim 15 wherein the pre-known data are provided by a human operator.

17. The method of claim 15 wherein the pre-known data are provided by a previous application of the method.

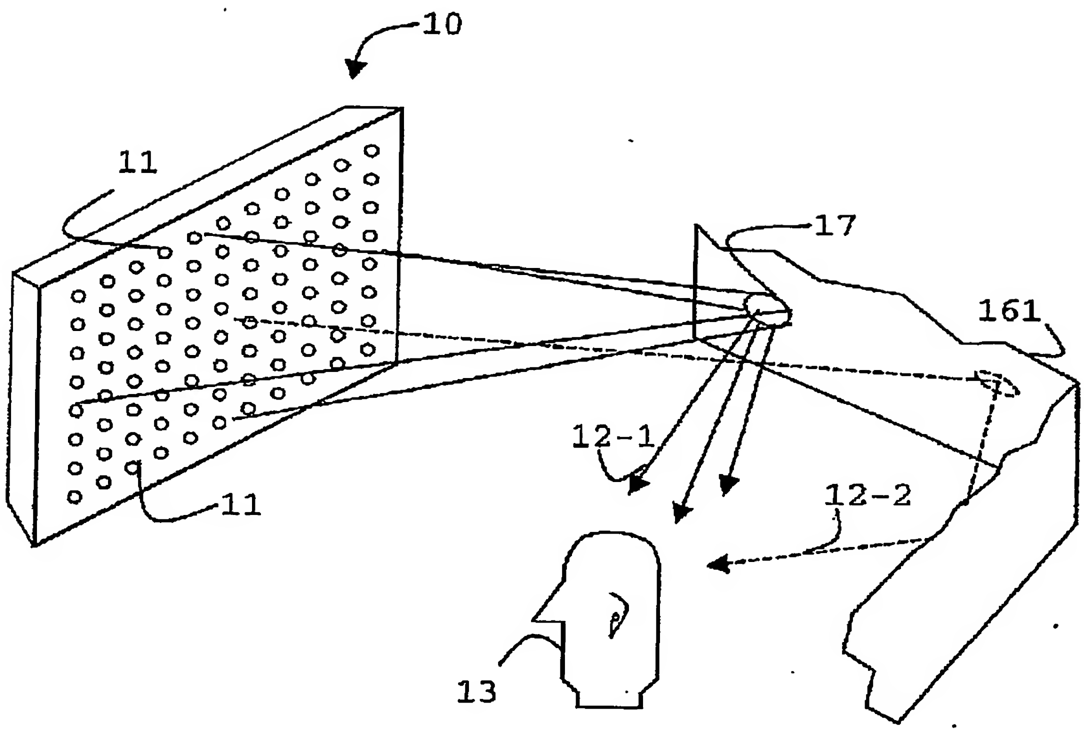


FIG. 1

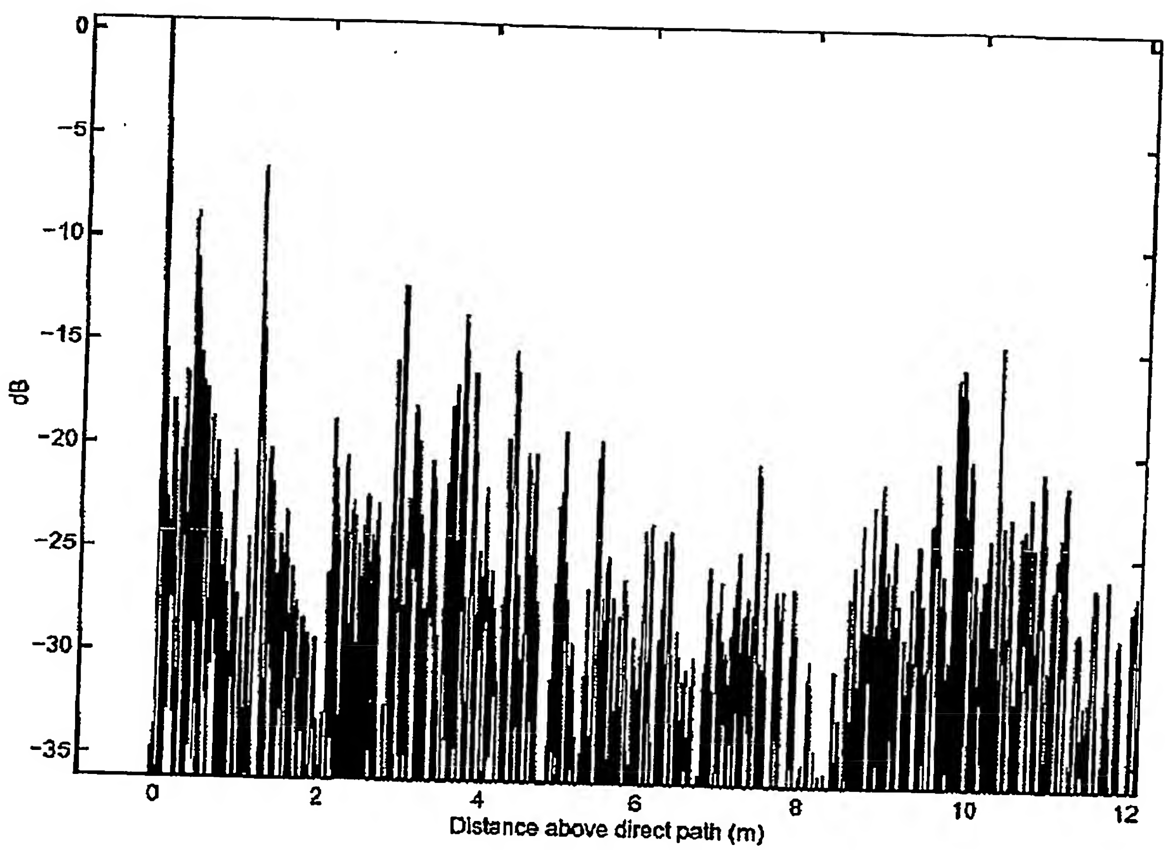


FIG. 2

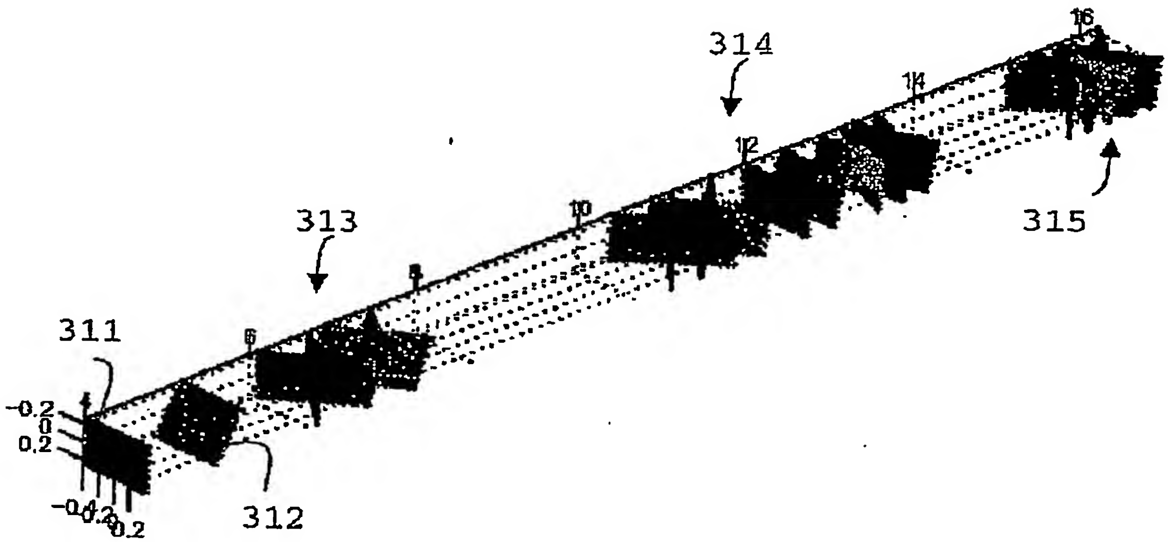


FIG. 3

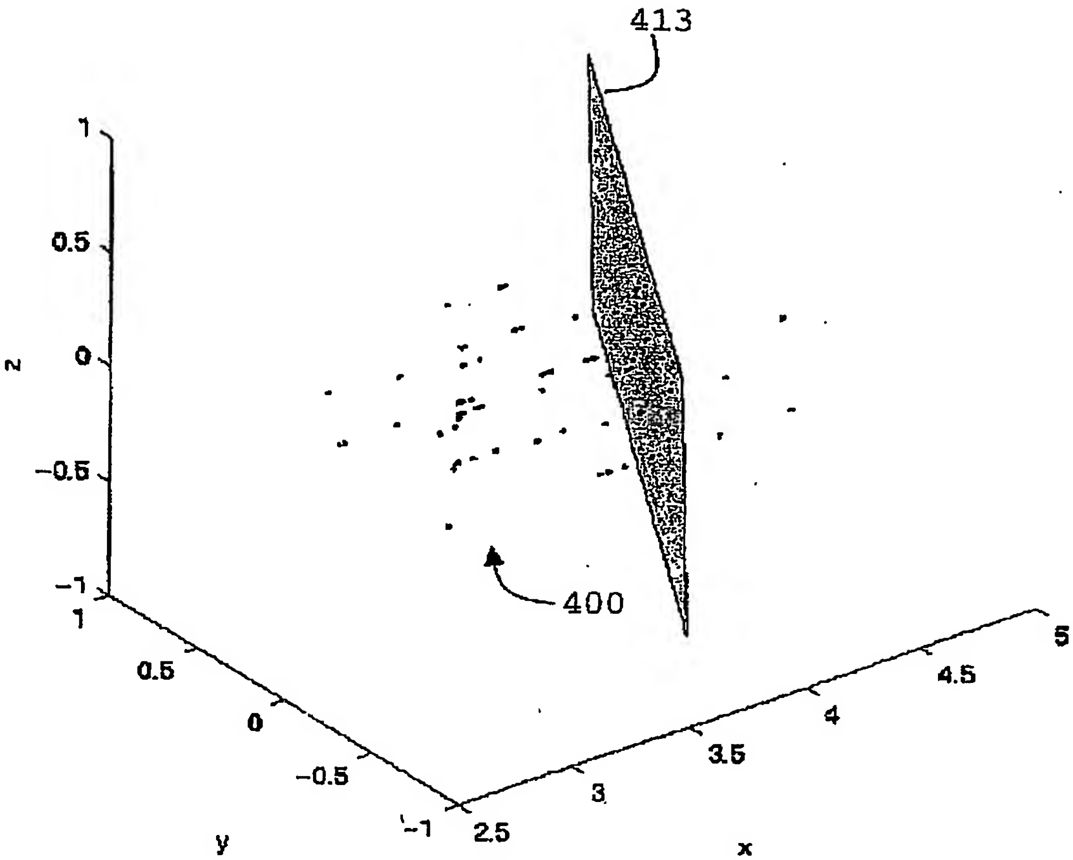


FIG. 4A

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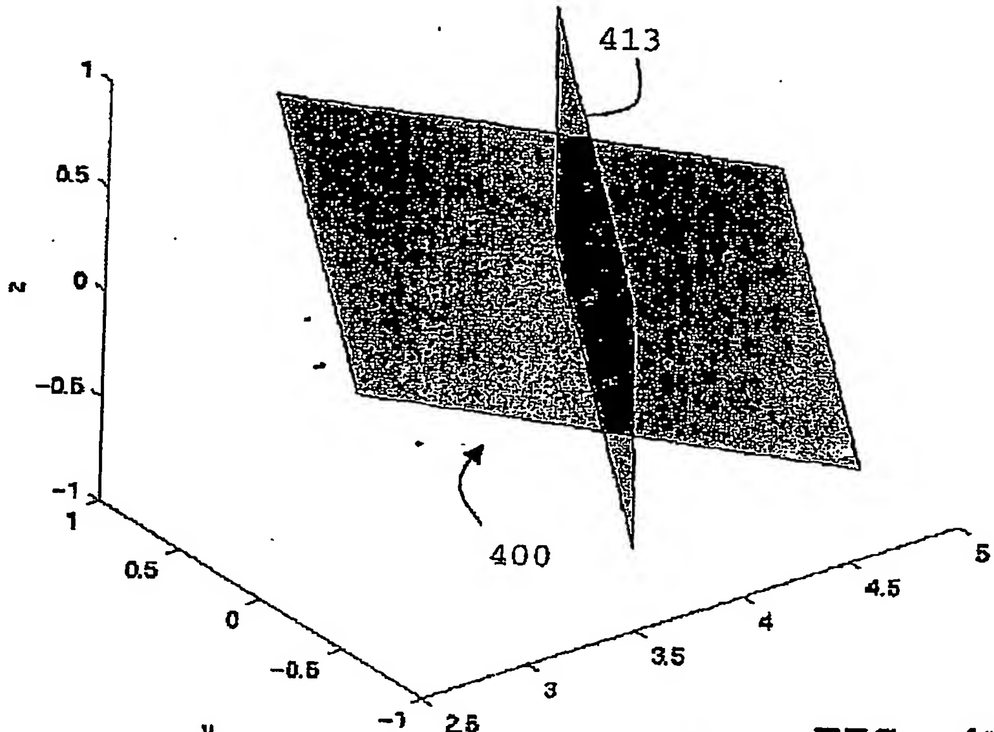


FIG. 4B

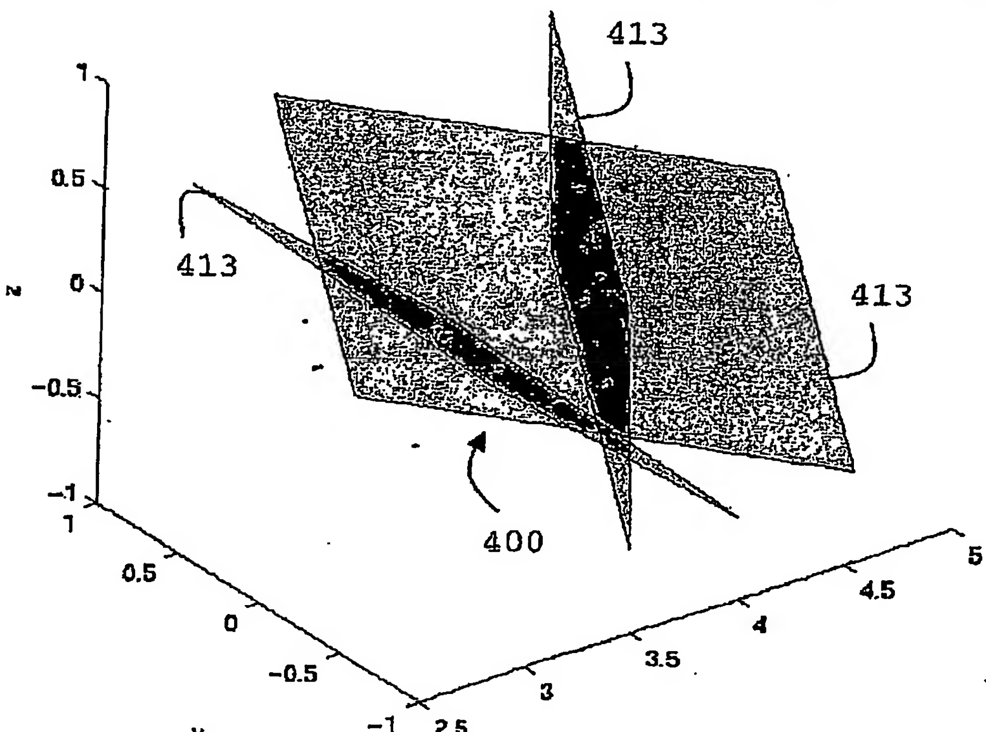


FIG. 4C

P163-GB



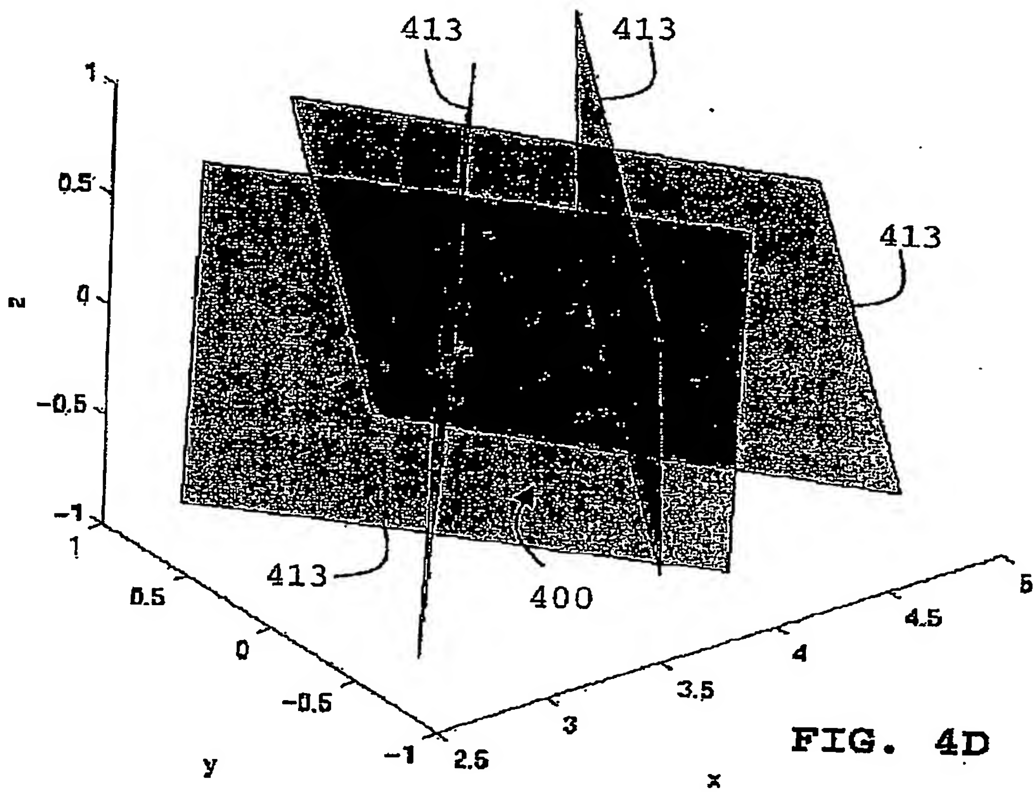


FIG. 4D

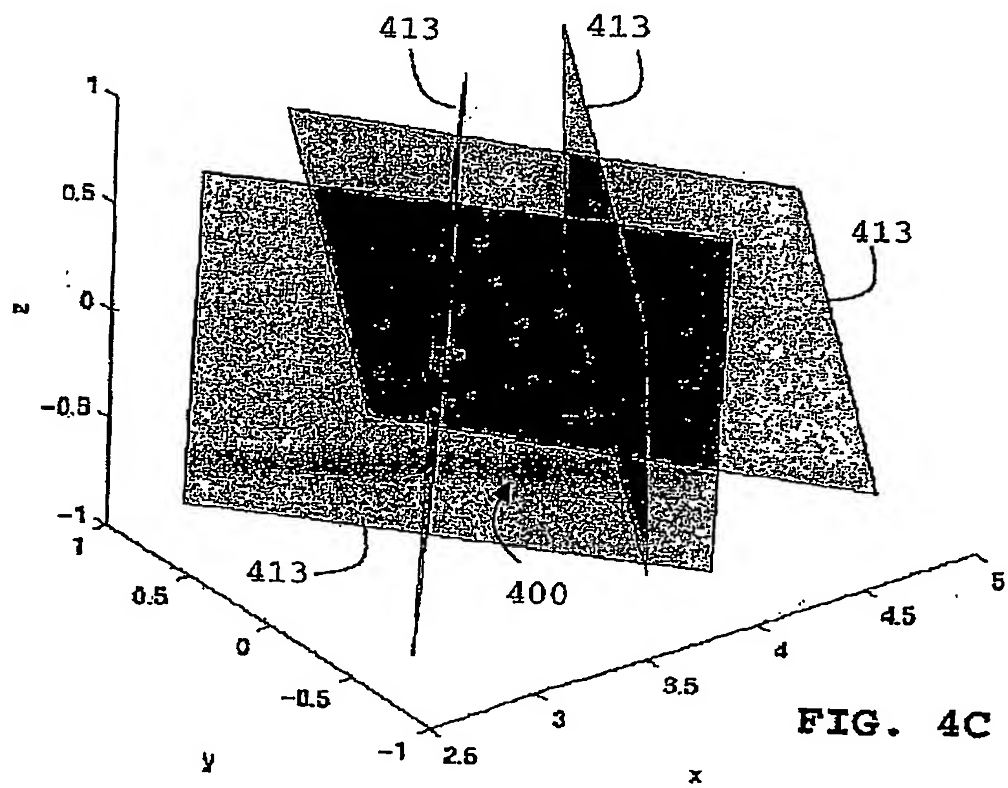


FIG. 4C

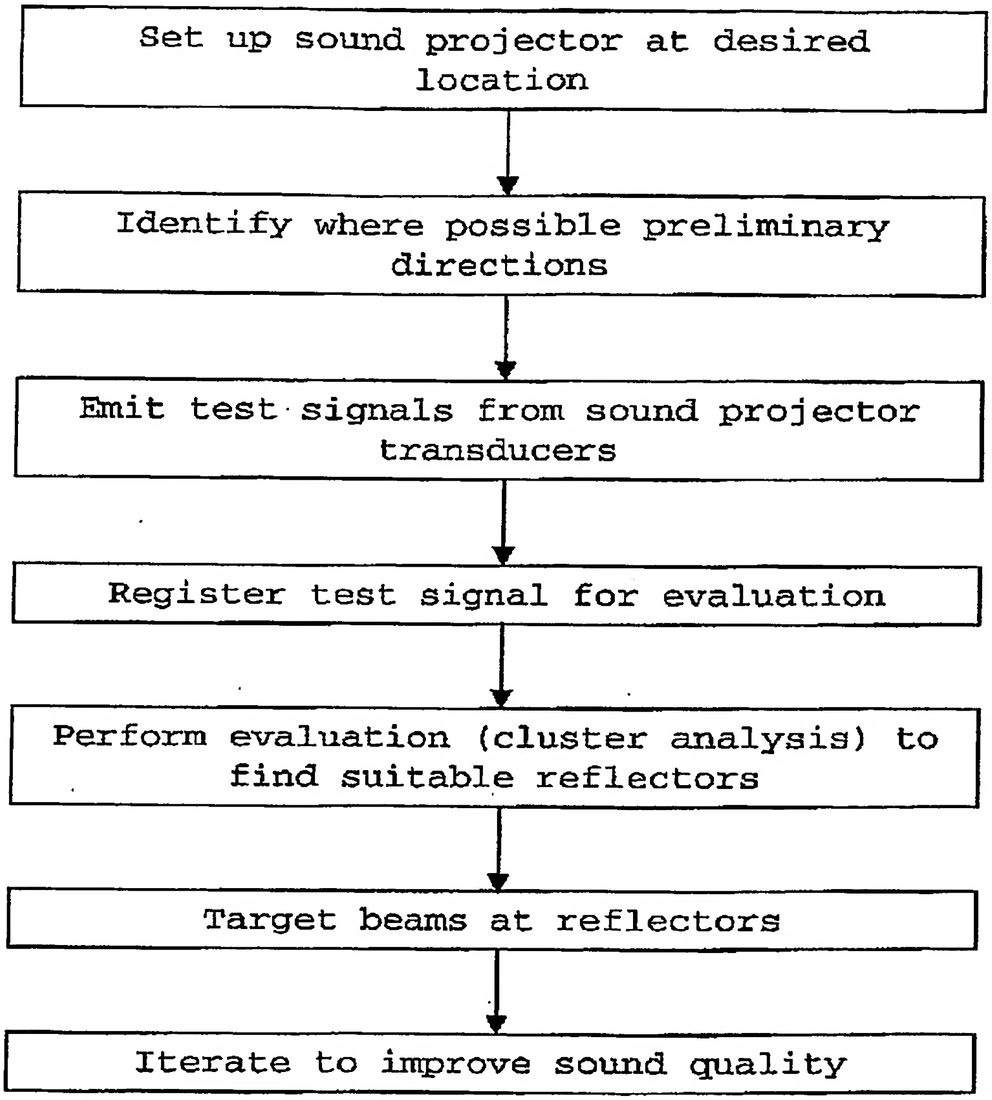
Set up sound projector at desired location
Identify where possible preliminary directions
Emit test signals from sound projector transducers
Register test signal for evaluation
Perform evaluation (cluster analysis) to find suitable reflectors
Target beams at reflectors
Iterate to improve sound quality

FIG. 5